## DETAILED ACTION

### *Examiner Comment*

1. The reference EP 0741471 has been considered by the Examiner and listed on attached PTO-892. The reference is not considered to affect the allowability of claims 1-23 of the instant application because EP 0741471 concerns measuring transmission quality through determining echo delay and does not consider/calculate the processing delay of a speech signal as in the instant claimed invention.

### *Allowable Subject Matter*

2. The following is an examiner's statement of reasons for allowance:

As per claim 1, the closest known prior art fails to teach or fairly suggest, alone or in reasonable combination:

A method for evaluating a processing delay of a speech signal contained in data packets received in a receiver terminal during a voice call to a terminal sending said data packets over a packet-switched network, the receiver terminal having a telephony module which generates a reconstituted speech signal from the received data packets, said method comprising the steps of:

obtaining, at the receiver terminal, a stream of audio packets from the received data packets and decoding the audio packet stream within a predetermined decoding time to reconstitute a first speech signal from the received packets of the audio stream;

duplicating, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module to constitute a second speech signal;

determining, at the receiver terminal, a time difference between the first speech signal and the second speech signal; and

calculating, at the receiver terminal, the processing delay of the speech signal contained in the data packets received in the receiver terminal from at least the determined time difference between said first and second speech signals and said predetermined decoding time.

Galetto et al. teaches the determination of speech latency across a communication network element having an input interface and an output interface includes allocating a timestamp to the data packets of a sample of data packets representing a speech signal at the two interfaces, recording the timestamps together with the corresponding data packets, decoding the recorded data packets at both interfaces to generate respective envelopes in the time domain, cross-correlating the envelopes to determine correlating areas of the envelopes, and determining a value for the speech latency between the interfaces from the timestamps associated with correlating areas of the envelopes. (Abstract) Galetto, however, fails to teach the limitation of duplicating, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module to constitute a second speech signal. Galetto, therefore, also fails to teach the subsequent limitations involving the calculation of the

processing delay of the speech signal because the first and second speech signals are not both generated at the receiver terminal.

Psytechnics teaches a determination of end-to-end delay based on the additional information about the delay estimate from RTCP packets, coding and packetization delay, jitter delay, and access delay from both the send and receive side. (Pages 2-3) Psytechnics, however, fails to teach the limitation of duplicating, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module to constitute a second speech signal. Psytechnics, therefore, also fails to teach the subsequent limitations involving the calculation of the processing delay of the speech signal because the first and second speech signals are not both generated at the receiver terminal.

Kirla teaches the calculation of the transmission delay of a packet-switched network by using a Ping technique. (col. 8, lines 36-40) Kirla, however, fails to teach the limitation of duplicating, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module to constitute a second speech signal. Kirla, therefore, also fails to teach the subsequent limitations involving the calculation of the processing delay of the speech signal because the first and second speech signals are not both generated at the receiver terminal.

Schaffer teaches sending end-to-end delay information over a packet-switched network to a collection server configured to manage end-to-end delay information sent by a plurality of communication terminals connected to a network. (col. 2, lines 32-35) Schaffer, however, fails to teach the limitation of duplicating, at the receiver terminal, at

least a portion of the speech signal reconstituted by the telephony module to constitute a second speech signal. Schaffer, therefore, also fails to teach the subsequent limitations involving the calculation of the processing delay of the speech signal because the first and second speech signals are not both generated at the receiver terminal.

Claims 2-13 are also considered allowable for depending on, and further limiting, allowable claim 1.

As per claim 14, the closest known prior art fails to teach or fairly suggest, alone or in reasonable combination,

A device for evaluating a processing delay of a speech signal contained in data packets received in a receiver terminal during a voice call to a terminal sending said data packets over a packet-switched network, the receiver terminal having a telephony module which generates a reconstituted speech signal from the received data packets, said device comprising:

a network filter module configured to obtain, at the receiver terminal, a stream of audio packets from the received data packets;

a control decoder module having a predetermined decoding time for decoding the stream of audio packets obtained and for reconstituting a first speech signal from the received packets of the audio stream;

an audio filter module configured to duplicate, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module, the duplicated portion of the speech signal constituting a second speech signal;

means for determining, at the receiver terminal, a time difference between the first speech signal and the second speech signal; and

means for calculating, at the receiver terminal, the processing delay of the speech signal contained in data packets received in the receiver terminal from at least the determined time difference between said first and second speech signals and the predetermined decoding time.

Galetto et al. teaches the determination of speech latency across a communication network element having an input interface and an output interface includes allocating a timestamp to the data packets of a sample of data packets representing a speech signal at the two interfaces, recording the timestamps together with the corresponding data packets, decoding the recorded data packets at both interfaces to generate respective envelopes in the time domain, cross-correlating the envelopes to determine correlating areas of the envelopes, and determining a value for the speech latency between the interfaces from the timestamps associated with correlating areas of the envelopes. (Abstract) Galetto, however, an audio filter module configured to duplicate, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module, the duplicated portion of the speech signal constituting a second speech signal. Galetto, therefore, also fails to teach the

subsequent limitations involving the calculation of the processing delay of the speech signal because the first and second speech signals are not both generated at the receiver terminal.

Psytechnics teaches a determination of end-to-end delay based on the additional information about the delay estimate from RTCP packets, coding and packetization delay, jitter delay, and access delay from both the send and receive side. (Pages 2-3) Psytechnics, however, an audio filter module configured to duplicate, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module, the duplicated portion of the speech signal constituting a second speech signal. Psytechnics, therefore, also fails to teach the subsequent limitations involving the calculation of the processing delay of the speech signal because the first and second speech signals are not both generated at the receiver terminal.

Kirla teaches the calculation of the transmission delay of a packet-switched network by using a Ping technique. (col. 8, lines 36-40) Kirla, however, an audio filter module configured to duplicate, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module, the duplicated portion of the speech signal constituting a second speech signal. Kirla, therefore, also fails to teach the subsequent limitations involving the calculation of the processing delay of the speech signal because the first and second speech signals are not both generated at the receiver terminal.

Schaffer teaches sending end-to-end delay information over a packet-switched network to a collection server configured to manage end-to-end delay information sent

by a plurality of communication terminals connected to a network. (col. 2, lines 32-35) Schaffer, however, an audio filter module configured to duplicate, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module, the duplicated portion of the speech signal constituting a second speech signal. Schaffer, therefore, also fails to teach the subsequent limitations involving the calculation of the processing delay of the speech signal because the first and second speech signals are not both generated at the receiver terminal.

Claims 15-18, 21-23 are also considered allowable for depending on, and further limiting, allowable claim 14.

As per claim 19, the closest known prior art fails to teach or fairly suggest, alone or in reasonable combination,

A computer-readable storage medium encoded with a computer program executed by a computer that causes evaluation of a processing delay of a speech signal contained in data packets received in a receiver terminal during a voice call to a terminal sending said data packets over a packet-switched network, the receiver terminal having a telephony module which generates a reconstituted speech signal from the received data packets, the computer program comprising:

program code for obtaining, at the receiver terminal, a stream of audio packets from the received data packets and decoding the audio packet stream within a predetermined decoding time to reconstitute a first speech signal from the received

packets of the audio stream;

program code for duplicating, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module to constitute a second speech signal;

program code for determining, at the receiver terminal, a time difference between the first speech signal and the second speech signal; and

program code for calculating, at the receiver terminal, the processing delay of the speech signal contained in the data packets received in the receiver terminal from at least the determined time difference between said first and second speech signals and said predetermined decoding time.

Galetto et al. teaches the determination of speech latency across a communication network element having an input interface and an output interface includes allocating a timestamp to the data packets of a sample of data packets representing a speech signal at the two interfaces, recording the timestamps together with the corresponding data packets, decoding the recorded data packets at both interfaces to generate respective envelopes in the time domain, cross-correlating the envelopes to determine correlating areas of the envelopes, and determining a value for the speech latency between the interfaces from the timestamps associated with correlating areas of the envelopes. (Abstract) Galetto, however, an audio filter module configured to duplicate, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module, the duplicated portion of the speech signal

constituting a second speech signal. Galetto, therefore, also fails to teach the subsequent limitations involving the calculation of the processing delay of the speech signal because the first and second speech signals are not both generated at the receiver terminal.

Psytechnics teaches a determination of end-to-end delay based on the additional information about the delay estimate from RTCP packets, coding and packetization delay, jitter delay, and access delay from both the send and receive side. (Pages 2-3) Psytechnics, however, an audio filter module configured to duplicate, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module, the duplicated portion of the speech signal constituting a second speech signal. Psytechnics, therefore, also fails to teach the subsequent limitations involving the calculation of the processing delay of the speech signal because the first and second speech signals are not both generated at the receiver terminal.

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Schaffer teaches sending end-to-end delay information over a packet-switched network to a collection server configured to manage end-to-end delay information sent by a plurality of communication terminals connected to a network. (col. 2, lines 32-35) Schaffer, however, an audio filter module configured to duplicate, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module, the duplicated portion of the speech signal constituting a second speech signal. Schaffer, therefore, also fails to teach the subsequent limitations involving the calculation of the processing delay of the speech signal because the first and second speech signals are not both generated at the receiver terminal.

As per claim 20, the closest known prior art fails to teach or fairly suggest, alone or in reasonable combination,

A computer-readable storage medium encoded with a computer program executed by a computer that causes evaluation of a processing delay of a speech signal contained in data packets received in a receiver terminal during a voice call to a terminal sending said data packets over a packet-switched network, the receiver terminal having a telephony module which generates a reconstituted speech signal from the received data packets, the computer program comprising:

program code for obtaining, at the receiver terminal, a stream of audio packets from the received data packets and decoding the audio packet stream within a predetermined decoding time to reconstitute a first speech signal from the received packets of the audio stream;

program code for duplicating, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module to constitute a second speech signal;

program code for determining, at the receiver terminal, a time difference between the first speech signal and the second speech signal; and

program code for calculating, at the receiver terminal, the processing delay of the speech signal contained in the data packets received in the receiver terminal from at least the determined time difference between said first and second speech signals and said predetermined decoding time; and

program code for evaluating the calculated processing delay of the speech signal in the terminal to evaluate end-to-end transmission delay of the speech signal contained in the data packets received in the receiver terminal during the voice call to the receiver terminal sending said speech signal over the packet-switched network.

Galetto et al. teaches the determination of speech latency across a communication network element having an input interface and an output interface includes allocating a timestamp to the data packets of a sample of data packets representing a speech signal at the two interfaces, recording the timestamps together with the corresponding data packets, decoding the recorded data packets at both interfaces to generate respective envelopes in the time domain, cross-correlating the envelopes to determine correlating areas of the envelopes, and determining a value for the speech latency between the interfaces from the timestamps associated with

correlating areas of the envelopes. (Abstract) Galetto, however, an audio filter module configured to duplicate, at the receiver terminal, at least a portion of the speech signal reconstituted by the telephony module, the duplicated portion of the speech signal constituting a second speech signal. Galetto, therefore, also fails to teach the subsequent limitations involving the calculation of the processing delay of the speech signal because the first and second speech signals are not both generated at the receiver terminal.

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signal because the first and second speech signals are not both generated at the receiver terminal.

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3. Any comments considered necessary by applicant must be submitted no later than the payment of the issue fee and, to avoid processing delays, should preferably accompany the issue fee. Such submissions should be clearly labeled “Comments on Statement of Reasons for Allowance.”

## ***Conclusion***

4. Any inquiry concerning this communication or earlier communications from the examiner should be directed to GREG A. BORSETTI whose telephone number is (571)270-3885. The examiner can normally be reached on Monday - Thursday (8am - 5pm Eastern Time).

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, RICHEMOND DORVIL can be reached on 571-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/Greg A. Borsetti/
Examiner, Art Unit 2626

/Talivaldis Ivars Smits/
Primary Examiner, Art Unit 2626 4/20/2010